RESEARCH ARTICLE OPEN ACCESS

Reducing latency in Voice over Internet Protocol Using Periority Queuing Technique

Amaechi, Chijindu O*, Dr. Michael.O. Alor**, Ezema, Christopher Ikechukwu***, Ozo, Victor chika , ****
Electrical and Electronics Engineering Department, Enugu State University of Science and Technology, ESUT, Enugu
Email; jinduamaechi@yahoo.com

--************************----------------------------------

Abstract:

This work presents reducing latency in voice over internet protocol using priority queuing technique. Over the internet protocol (IP) network, voice over internet protocol (VoIP) packets compete with other data types (ftp, email, video, etc) for the limited network resources, resulting in congestion, packet loss and high latency to the delay-sensitive VoIP packets. This has been a major setback to the achievement of satisfactory user experience for VoIP over the years. In order to reduce the latency in VoIP, this work has used Priority Queuing technique, an algorithm that has been utilized to accord high priority to the VoIP packets over every other traffic type. In this work, a live VoIP network has been first characterized to determine the effect of latency on VoIP. It was observed that as the number of VoIP packets sent increased, the latency also increased, while there were increasing packet losses (resulting in progressive reduced throughput percentage). To significantly improve on this situation, the same network has been modelled using Simulink. Priority Queuing algorithm was applied on the modelled network to accord high priority to the voice packets over every other packet type, and the results were tabulated. The result has shown that the latency for VoIP reduced by 50.47%, while the throughput percentage improved by 30%.By this project, the delay in VoIP has been significantly reduced and the quality of service greatly enhanced.

Keywords **—Latency, Congestion, VoIP, Quality of Service, an d Throughput.**

--************************----------------------------------

I. INTRODUCTION

Voice over internet protocol (VoIP)is the method of transmitting voice over an internet protocol (IP) network. It describes telephony devices that use IP as the native transport for voice traffic.In VoIP, the analog audio signals sampled are encoded into the digital communication stream, encapsulated into IP packets and then transmitted over the internet together with other traffic types like video, e-mail, web data, file transfer protocol (FTP), etc.Therefore, as a widely accepted alternative of traditional telephone service, VoIP provides voice communication services using IP network.

Since the same network is used to simultaneously transport voice and all the other packet types, there is competition for network resources among the various traffic types. This leads to network congestion, delay and packets loss as the packets are buffered in waiting queues before being transmitted across the network. This delay is referred to as latency. Because VoIP is an interactive and real-time application, it is very delay-sensitive as its quality of service (QoS) is negatively impacted once the delays exceed 150miliseconds.

Queuing is a congestion management technique that keeps the incoming packets in queues, and governs how they are buffered while waiting to be transmitted. For the VoIP packets to experience minimal delay during transmission and deliver good QoS, there is a need to devise a technique that accords priority to the voice packets over the other non-real time packets like e-mail and FTP.

This work proposes to solve this problem using Priority Queuing (PQ) technique with the hope of improving quality of VoIP by managing network traffic based on its priority and thus reduce latency. The working mechanism of PQ technique is explained in the next section (1.2)

Statement of the Problem

The main problem is how to reduce latency so as to improve the Quality of Service (QoS) of such a delay-sensitive application as the voice over internet protocol (VoIP).

For a network that is designed to support different traffic types (mails, files, voice, video, etc) which share a common path between routers, there is competition for network resources among the different traffic types, leading to congestion. Network delay (latency) is one of the consequences of congestion. However, voice over internet protocol, being a real time application, has an extreme bandwidth and delay-sensitivity, which adversely affects its quality.

In order to solve the problem, (i) the situation should be properly understood and analysed, (ii) there is the need to devise a technique that accords priority to VoIP traffic over the other less delay-sensitive traffic types, (iii) a post-mortem analysis should be carried out in order to determine the impact of the applied solution in solving the problem.

Solving this problem of latency in VoIP is very imperative to deliver excellent quality of service to the end users of VoIP. It is also necessary to allow users of VoIP communicate over the internet services at low cost.

Aim and Objectives of the Study

The aim of this research is reducing latency in voice over internet protocol using priority queuing technique.

The objectives are:

i. To evaluate the operational parameters of the network under study and determine the effect of latency(delay) on voice over internet protocol
ii. To develop a model that will reduce the delay using Priority Queuing technique
iii. To implement the model using Simulink
iv. To simulate the model and determine its impact on the network

Significance of the Study

The demand for quality service in both wired and wireless networks has grown rapidly over recent years especially within the networks of larger companies and organizations. This research work will identify delay on voice over internet protocol (VoIP) and find a suitable mechanism to reduce the delay and improve the quality of service thereby guaranteeing customer satisfaction.

Scope of the Study

This study will be conducted using priority queuing technique to reduce latency in the VoIP of a wireless Wide Area Network (WAN) in which subsidiary Local Area Networks (LAN) are wirelessly interconnected via an internet cloud. The result shall be assessedthrough reduced delay time for packet arrival and improved throughput.

II. MATERIAL AND METHODS

Materials Used

The case study network setup is designed using active queuing management scheme and presented using a physical setup consisting of switches, wireless access points, Personal Computers, IP phones, RJ45 cables, Ethernet adapter, Linux server, PC with installed network monitor software (Asterisk), ISDN phones interconnected to the cloud based on IEEE 802.11 standard.

The modelled system was implemented using the personal computer, signal processing toolbox, communication toolbox, optimization toolbox and wireless local area network toolbox of Simulink.

METHODS

Evaluation of the Operational Parameters of the Network under Study to Determine the Effect of Latency (delay) on Voice over Internet Protocol

This work characterizes the real-life wireless VoIP network of the Bolsa Technologies. The characterization was done to evaluate the operational parameters of the network and see effect of latency on VoIP. The network data was collected using real time data emitter method, since it provides more reliable and accurate data compared to analytical or network simulation methods of characterization. The real time data emitter method employs physical equipment to investigate the network configuration of various parameters on different network deployments, communication standards and protocols.

The user equipment was used to communicate voice, http and FTP data while the latency performance of VoIP (our traffic of interest) was measured. The service quality was graded based on Mean Opinion Score (MOS) rating defined by ITU-T as a 5-point rating scale shown in Table 1 below:

Table 1: Mean Opinion Score (MOS) Classification for Wireless Network

Point	Throughput (%)	MOS
5	90 and above	Excellent
4	89 -75	Good
3	74- 65	Fair
2	64-55	Poor
1	54 and below	Bad

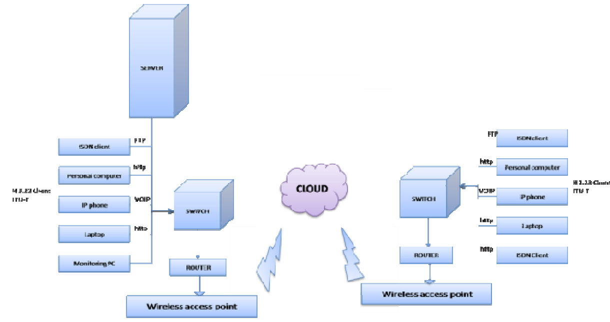


Figure 1: The Diagram of Bolsa Technologies VoIP Network (Source: Bolsa Technologies.)

From the setup above in Figure 1, the wireless communications network of multiple users was presented. The users performed end to end communication of packets like the http, FTP and VoIP via the H 3.23 clients over the same network. The monitoring PC installed with Asterisk software was connected to one end of the network with the aim of measuring the number of VoIP, FTP & HTTP packets transmitted from one end, the number of packets received at the other end, and the delivery time (latency) of the VoIP packets (our packets of interest). The results were presented as shown below in Table .2

Table 2: Network Performance (characterized) Showing Competition among Different Packet Types

Packets Sent (Kbits)			Packets received (Kbits)			Latency of VoIP Packets (ms)	Throughput of VoIP Packets (%)	MOS Rating for VoIP
VoIP	FTP	HTTP	VoIP	FTP	HTTP			
1000	1000	1000	980	1000	990	40	98	Excellent
1200	1200	1200	1070	1080	1140	45	89	Good
1300	1300	1300	1106	1170	1144	49	85	Good
1400	1400	1400	1111	1260	1120	57	79	Good
1500	1500	1500	1125	1275	1275	79	75	Fair
1600	1600	1600	1137	1440	1424	126	71	Fair
1700	1700	1700	1139	1530	1479	149	67	Fair
1800	1800	1800	1170	1584	1512	159	65	Poor
1900	1900	1900	1198	1672	1710	168	63	Poor
2000	2000	2000	1201	1740	1760	180	60	Poor
Average						**105.2**	**75.2**	

The network was analyzed as shown using the graph in Figure 2.

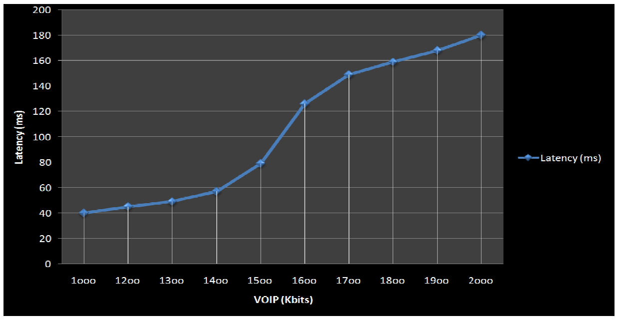


Figure 2: Network Latency Against VoIP Packets Sent

From the result presented in Figure 2, it was observed that the latency increases as the number of VoIP packets sent increased. This is because the VoIP packets are competing for network resources with the other service types (FTP, http). The VoIP data being delay sensitive experiences severe packet drops (as tabulated in the percentage of throughput). This effect is analyzed as shown in the graph of Figure 3 below:

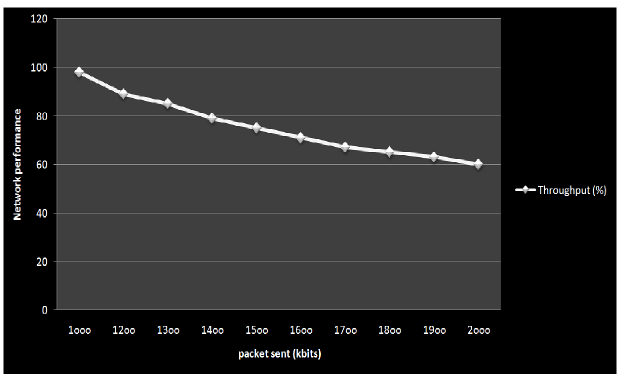


Figure 3: Network Performance Against Amount of VoIP Packets Sent

The graph reveals that the quantity of VoIP data delivered (throughput %) reduced as the number of packets sent increased. This is because of the lack of intelligent congestion management scheme in the network. Hence there is need for an intelligent queuing mechanism which can give higher priority to delay sensitive traffic like VoIP.

III. DEVELOPMENT OF A MODEL TO REDUCE THE LATENCY IN VOIP USING PRIORITY QUEUING TECHNIQUE

The packets queue has been modelled based on **Little's Law** which states that the average number of packets (E[N]) in a queue is the product of the average arrival rate of packets (λ) and the average length of time (E[D]) it spends in the queue. Expressed mathematically,

$$E[N] = E[D] * \lambda$$

The Proposed System

The proposed system will be developed to optimize the voice over internet performance of the characterized network using priority queuing technique. The proposed block diagram is presented below.

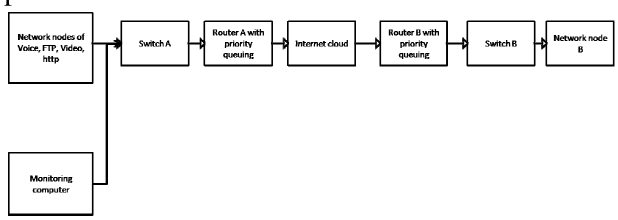


Figure 4: Block Diagram of the Proposed System with Priority Queuing

From the block diagram in Figure 4 above, the sender end consists of network nodes (which can transmit various types of packets), monitoring computer, Switch A and Router A. The receiver end consists of Router B, Switch B, and network nodes. Both ends of the network are linked via the Internet cloud. Router A and Router B are programmed for Priority Queuing, according high priority to VoIP packets. The monitoring computer evaluates the network performance for data analysis.

Development of the Priority Queuing Technique

Priority queuing (PQ) is the basis for a class of queue scheduling algorithms that are designed to provide a relatively simple method of supporting differentiated service classes. In this method of queuing, data are classified based on high with classification label of “0.5”, low with classification label of “-0.5” or medium priority with classification label of “0” based on the packet type and then queued for throughput. Packets are scheduled from the head of a given queue only if all queues of higher priority are empty. Within each of the priority queues, packets are scheduled in First-In First-Out (FIFO) order as shown in Figure 5.

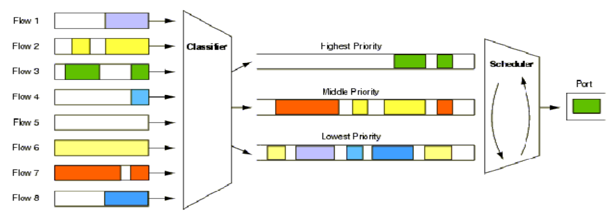


Figure 5: Schematic Diagram of Priority Queueing Algorithm

PQ offers a couple of benefits: For software-based routers, PQ places a relatively low computational load on the system when compared with more elaborate queuing disciplines. PQ allows routers to organize buffered packets, and then service one class of traffic differently from other classes of traffic. The priorities are set so that real time applications such as interactive voice and video get priority over applications that do not operate in real time. This process enables undisturbed high priority traffic to travel through the network with much lower latency compared to the low priority traffic. The algorithmic flow chart is presented below;

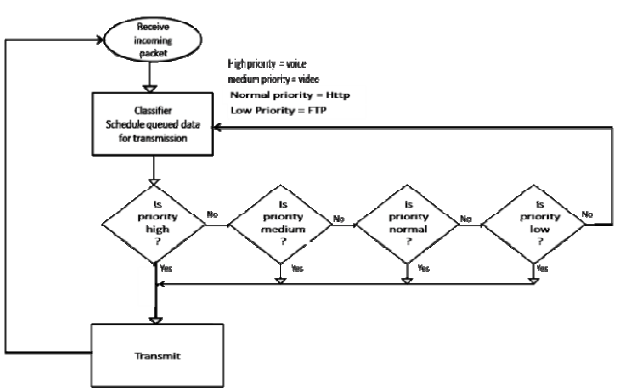


Figure 6: Flow Chart of the Priority Queuing Algorithm

In the flowchart of Figure 6, the incoming packets are queued by the classifier according to the traffic types into queues 1, 2, 3 and 4 to represent voice, video, email, FTP etc being transmitted over the network. At the scheduler, priorities are assigned. Here, voice (Queue 1) is assigned High, while video, email, and FTP are assigned Medium, Normal and Low respectively. The algorithm is such that a traffic type of higher level of priority must be transmitted before any one of lower level receives attention.

The model network diagram to reduce delay in VoIP using priority queuing technique is presented in Figure 7 below:

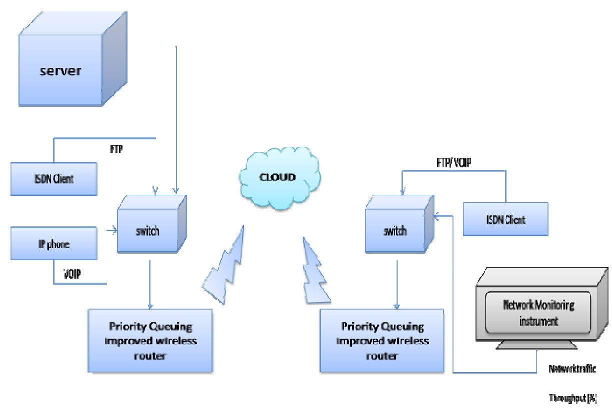
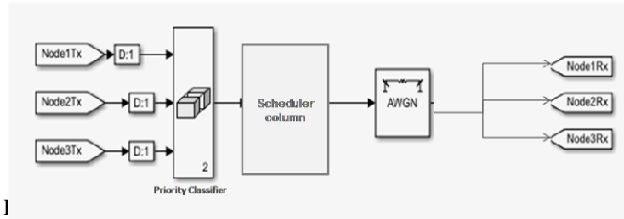


Figure 7: The Model Network Diagram for Reduction of Delay in VoIP using PQ

Implementation of the Model Using Simulink

The model system was implemented using the signal processing toolbox, communication toolbox, optimization toolbox, wireless local area network toolbox and Simulink.

The proposed network structure employs three nodes which transmit various packets to the other end of the network. These packets transmitted from each node over the controlled channel are managed in the router by PQ as shown below in Figure 8.



From the Simulink model in Figure 8, the three network nodes transmitting packets (FTP and VoIP) are identified by the priority classifier in the network and queued by the scheduler according the type of packet transmitted. The signal with the highest priority is scheduled for throughput to the designated receiver node. The Simulink model of the wireless network consisting of the three nodes is presented below in Figure 9.

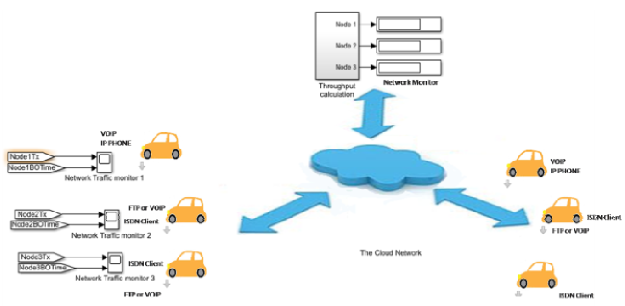


Figure 9: The Simulated Network

From the model presented in Figure 9, six network nodes were employed for the system design and simulation process. Three nodes were used at each end of the network. The nodes used to set the network are the four ISDN clients and two IP phones. The routing process for each node was controlled using the priority queuing model developed, with the performance of the network measured. The source code for the implementation of the Simulink model was presented in Appendix A.

IV. RESULTS AND DISCUSSION

Here presents the result of the models simulated and the effect of the priority queuing on the network management to reduce latency and congestion. From the model in Figure 9 (see chapter three), the three network nodes located at one end of the network were each used to communicate with other three respective nodes at the other end of the network.

The result in Table 4.1 below shows that priority has been given to VoIP packets over the other packet types (FTP & HTTP), resulting into a very meaningful improvement in the quality of the network with respect to reduced latency and excellent throughput percentage for VoIP packets.

Table 3 Network Performance (Improved Network) Showing Priority for VoIP Packets Over Other Packet Types

Packets Sent (Kbits)			Packets received (Kbits)			Latency of VoIP Packets (ms)	Throughput of VoIP Packets (%)
VoIP	FTP	HTTP	VoIP	FTP	HTTP		
1000	1000	1000	980	900	910	40	98
1200	1200	1200	1140	810	850	45	95
1300	1300	1300	1261	799	810	45	97
1400	1400	1400	1372	640	685	48	98
1500	1500	1500	1440	622	637	53	96
1600	1600	1600	1568	573	618	54	98
1700	1700	1700	1649	513	562	54	97
1800	1800	1800	1764	472	509	58	98
1900	1900	1900	1805	414	463	61	95
2000	2000	2000	1960	368	402	63	98
Average						**52.1**	**97**

In Table 4 below, the latencies of the characterized and improved systems have been compared. This is also depicted in the graph of Figure 10. From the result it can be observed that the new system has minimized latency to a very acceptable level for seamless transmission of voice traffic.

Table 4 Comparative Latency Performance for VoIP Packets

VOIP	Latency of Improved System (ms)	Latency of Characterized System (ms)
1000	40	40
1200	45	45
1300	45	49
1400	48	57
1500	53	79
1600	54	126
1700	54	149
1800	58	159
1900	61	168
2000	63	180

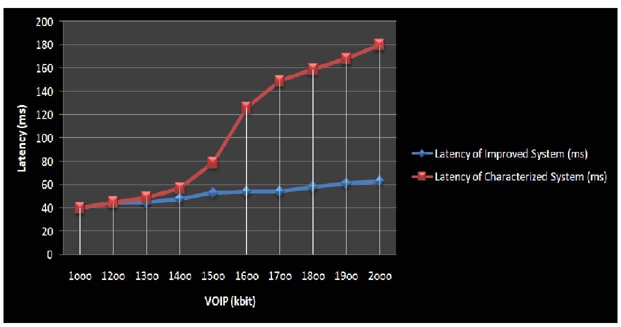


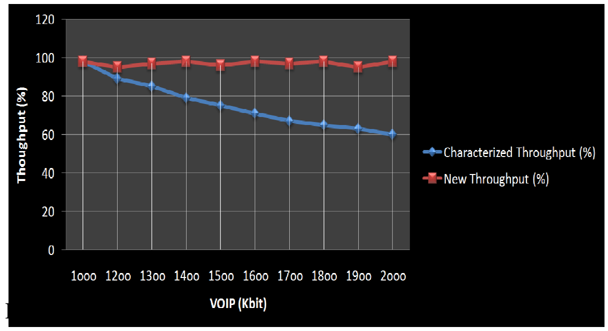
Figure 10: Comparative Performance for Latency

The comparative Throughput performance was also presented from the data of the characterized and new systems recorded in the new network and the result presented in Table 5 below.

Table 5: Comparative Throughput Performance for VoIP Packets

VOIP (Kbit)	Characterized Throughput (%)	New Throughput (%)
1000	98	98
1200	89	95
1300	85	97
1400	79	98
1500	75	96
1600	71	98
1700	67	97
1800	65	98
1900	63	95
2000	60	98

The result obtained in Table 5 is depicted in the graph of Figure 11 below. These show that the new throughput averaged at 97%.



V. CONCLUSIONS AND RECOMMENDATION

Contribution to Knowledge

This research work has successfully shown that priority queueing algorithm, apart from being a network queuing congestion management technique, has been successfully applied to reduce latency in voice over internet protocol (VoIP). This has therefore become a very effective technique to improve the quality of service (QoS) for VoIP user experience.

Conclusion

The rate of packets arrival for onward transmission to their respective destinations is very high when compared to the shared network resources like queue buffer, router memory and outgoing bandwidth. When congestion is experienced in the network, latency is experienced by numerous data packets and several packets get dropped as the queue starts overflowing. Excessive congestion in the network leads to throughput degradation and increased rate of packet loss. Network efficiency and reliability also reduce due to congestion and latency. The Active Queue Management (AQM)algorithms like random early detection (RED) and control delay (CoDel) help to check and react to the congestion prone situation, but they lack service differentiation capability to infer whether the traffic is deadline sensitive or not. Therefore, a queuing policy that implements easy service differentiation and accurate prioritization helps to build a stable transmission rate for delay sensitive data even at the time of congestion.

Recommendation

Having completed this research work, the author recommends the deployment of priority queuing technique by both private and public enterprise VoIP users to reduce delays so as to maximise their use of IP networks in delivery of voice services.

ACKNOWLEDGMENT

First, I thank the Almighty God for giving me the grace to successfully pursue and achieve this noble dream of obtaining the M.Eng. Degree from Enugu State University of Science and Technology

I specially thank my project supervisor Dr. Michael. O. Alor of the Department of Electrical and Electronics Engineering, ESUT, for his selfless guidance, support, suggestions, useful comments and constructive criticisms which all saw me through to the successful completion of this research work. Also, I thank all my wonderful lecturers and classmates for making the M. Eng. programme such a huge success story. In particular, I appreciate my personal friends in the class Engr. Christopher Iyke Ezema and Engr. Victor Chika Ozor. May God bless you all

Also, I thank the management of Bolsa Technologies for providing me with the real-life network data which was used at the network characterization stage of this research work.

Most prominently, I appreciate my beautiful wife Mrs. Vanessa Ijeoma Amaechi and our lovely children Judith, Mirabel and Duke for their prayers, strong support, encouragement and understanding all through the two years that this programme lasted. I love you all dearly.

Finally, I posthumously appreciate my beloved parents Late Chief Elijah Ogbonna Amaechi and lateEzinne Salome Enyidiya Amaechi for the wonderful sacrifices, impacts and investmentsthey made in my life. May God grant eternal rest to their gentle souls, Amen

REFERENCES

[1] Badr, S. and Darwesh, G., (2011). *QoS Adaption in Real Time Systems Based on CBWFQ*. 28th National Radio Science Conference (NRSC 2011), National Telecommunication Institute, Cairo, Egypt

[2] Balasundaram K, Velmurugan T, and Suresh R., (2014). Performance Analysis of Queuing Disciplines for Difference Services Using

[3] OPNET Tool, International Journal of Scientific Engineering and Technology (ISSN: 2277-1581), 3 (1), 1-4.

[4] Bevilacqua D., Masi M., Fischer M.J. and Garbin D.A., (2017).*Modeling The Performance of Low Latency Queueing for Emergency*. Telecommunications Proceedings of the 2017 Winter Simulation Conferences, Winipeg, Canada

[5] Brunonas, D., Tomas, A. and Aurelijus, B., (2016). *Analysis of QoS Assurance using Weighted Fair Queuing (WFQ) Scheduling Discipline with Low Latency Queue (LLQ).* 28th IEEE International Conference Information Technology Interfaces, Croatia, June 19-2016

[6] Bucheli S., (2014). Compensation Modeling for QoS Support on a Wireless Network. *Master degree thesis*.

[7] Dike J.N and Ani C.I (2014). Algorithmic Analysis of an Efficient Packet Scheduler for Optimizing the QoS of VoIP Networks, *International Journal of Computer Applications,* Volume 97, Number 4, pp5-12, retrieved from doi:10.5120/16993-7104 on June 2021.

[8] Gospindov, (2014).*The Effects of Different Queuing Disciplines over FTP, Video and VoIP Performance*, International Conference on Computer Systems & Technologies – CompSysTech'2014.

[9] Insaaf Mohamed, Morawaka M.D., Wanasinghe, W. A. I. N., Fernando, A. D., Mudugamuwa, M. A. D. R. and Dhammearatchi, D. (2016). Measuring the Queuing Delay in VoIP Packets, *COMPUSOFT, An international journal of advanced computer technology,* V (III), 1-5

[10] Kao M. (2015).Timing Jitter Control of an Add/drop Optical Module in a convergent Network, Wireless and Optical Communications,14th Annual WOCC International Conference

[11] Karem Rana, Khaled Menna, Atef Nadine and Alaa Hana, (2017), *Queuing Impact on VoIP QoS,* retrieved from www.researchgate.net/publication/314232147, on 28 March, 2020

[12] Malhotra N. (2013). Active Scheduling Algorithms in Congestion Management, A Review‖, *International Journal of Digital Application & Contemporary Research*, vol 1, issue no 8, 2013.

[13] Mohammed, Mirza Golam and Rashed. (2015).*A Comparative Study of Different Queuinging Techniques in VoIP,Video Conference and File Transfer*. Dept of ETE, Daffodil International University, pp28-30.

[14] Rashed M. M. and Kabir M. (2015). A comparative Study of Different Queuing Techniques in VoIP, Video Conferencing and File Transfer, *Daffodil International University Journal of Science and Technology*, Vol. 5, no. 1, pp. 37–47.

[15] Robert Luca, Petrica Ciotirnae and Florin Popescu, (2016). Influence of the QoS Measures for VoIP Traffic in a Congested Network, *International Journal of Computers Communications & Control ISSN 1841-9836*, 11(3), 5-9

[16] Sawaii Al. (2014). A Performance of Weighted Fair Queuing System with Multi-class Jobs, Advanced Information Networking and Application, IEEE International Conference, pp50-57.

[17] Semeria, C (2015). Supporting Differentiated Service Classes: Queue Scheduling Disciplines, White Paper, Juniper Networks, Inc.

[18] Yap K, A. Marshall and W. Yu. (2015).Providing QoS for Multimodal System Traffic Flows in Distributed Haptic Virtual Environments, Queen's University Belfast.

[19] Zheng L and Xu. (2011).Characteristics of Network Delay and Delay Jitter and its Effect on Voice over IP (VoIP) Communications, IEEE International Conference.

APPENDIX A: SOURCE CODES

```
%% Packetized Modem with six nodes
Declare node 1 and 4 as IP PHONES
Declare node 2 and 5 as ISDN CLIENTS
Declare node 3 and 6 as ISDN CLIENTS

runDuration = 10;   % Seconds
numPayloadBits = 19530;   % Bits
packetArrivalRate = 0.2;  % Packets per second
ackTimeOut = 0.25;        % ACK time out in
seconds
maxBackoffTime = 10;   % Maximum backoff time in
ackTimeOut durations
mMaxDataRetries = 5;   % Maximum DATA retries
queueSize = 10;   % Data Link Layer queue size in
packets
samplesPerFrame = 2000;   % Number of samples
processed every iteration
verbose = true;   % Print packet activity to
command line
input nodes number       % the transceivers in the
network
% Create packetized modem nodes
node1 and 4 = helperPacketizedModemNode(...
'Address', 1, ...
'DestinationList', [2, 3], ...
'NumPayloadBits', numPayloadBits, ...
'PacketArrivalRate', packetArrivalRate, ...
'ACKTimeOut', ackTimeOut, ...
'MaxBackoffTime', maxBackoffTime, ...
'MaxDataRetries', mMaxDataRetries, ...
'QueueSize', queueSize, ...
'CarrierDetectorThreshold', 1e-5, ...
'AGCMaxPowerGain', 65, ...
'SamplesPerFrame', samplesPerFrame, ...
'Verbose', verbose, ...
'SampleRate', sampleRate);
node2 and 5 = helperPacketizedModemNode(...
'Address', 2, ...
'DestinationList', [1 3], ...
'NumPayloadBits', numPayloadBits, ...
'PacketArrivalRate', packetArrivalRate, ...
'ACKTimeOut', ackTimeOut, ...
'MaxBackoffTime', maxBackoffTime, ...
'MaxDataRetries', mMaxDataRetries, ...
'QueueSize', queueSize, ...
'CarrierDetectorThreshold', 1e-5, ...
'AGCMaxPowerGain', 65, ...
'SamplesPerFrame', samplesPerFrame, ...
'Verbose', verbose, ...
'SampleRate', sampleRate);
node3 and 6 = helperPacketizedModemNode(...
'Address', 3, ...
'DestinationList', [1 2], ...
'NumPayloadBits', numPayloadBits, ...
'PacketArrivalRate', packetArrivalRate, ...
'ACKTimeOut', ackTimeOut, ...
'MaxBackoffTime', maxBackoffTime, ...
'MaxDataRetries', mMaxDataRetries, ...
'QueueSize', queueSize, ...
'CarrierDetectorThreshold', 1e-5, ...
'AGCMaxPowerGain', 65, ...
'SamplesPerFrame', samplesPerFrame, ...
'Verbose', verbose, ...
'SampleRate', sampleRate);

% Setup channel
channel = helperMultiUserChannel(...
'NumNodes', 3, ...
'EnableTimingSkew', true, ...
'DelayType', 'Triangle', ...
'TimingError', 20, ...
'EnableFrequencyOffset', true, ...
'PhaseOffset', 47, ...
'FrequencyOffset', 2000, ...
'EnableAWGN', true, ...
'EbNo', 25, ...
'BitsPerSymbol', 2, ...
'SamplesPerSymbol', 4, ...
'EnableRicianMultipath', true, ...
'PathDelays', [0
node1.SamplesPerSymbol/node1.SampleRate], ...
'AveragePathGains', [15 0], ...
'KFactor', 15, ...
'MaximumDopplerShift', 10, ...
'SampleRate', node1.SampleRate);

% Main loop
radioTime = 0;
nodeInfo = info(node1);
frameDuration =
node1.SamplesPerFrame/node1.SampleRate;
[rcvd1,rcvd2,rcvd3] =
deal(complex(zeros(node1.SamplesPerFrame,1)));
whileradioTime<runDuration
  trans1 = node1(rcvd1, radioTime);
  trans2 = node2(rcvd2, radioTime);
  trans3 = node3(rcvd3, radioTime);

% Multi-user channel
  [rcvd1,rcvd2,rcvd3] =
channel(trans1,trans2,trans3);

% Update radio time.
radioTime = radioTime + frameDuration;
end

%% Results
% Display statistics
nodeInfo(1) = info(node1);
nodeInfo(2) = info(node2);
nodeInfo(3) = info(node3);

for p=1:length(nodeInfo)
fprintf('\nNode %d:\n', p);
fprintf('\tNumGeneratedPackets: %d\n',
nodeInfo(p).NumGeneratedPackets)
fprintf('\tNumReceivedPackets: %d\n',
nodeInfo(p).NumReceivedPackets)
fprintf('\tAverageRetries: %f\n',
nodeInfo(p).Layer2.AverageRetries)
fprintf('\tAverageRoundTripTime: %f\n',
nodeInfo(p).Layer2.AverageRoundTripTime)
fprintf('\tNumDroppedPackets: %d\n',
nodeInfo(p).Layer2.NumDroppedPackets)
fprintf('\tNumDroppedPackets (Max retries): %d\n',
nodeInfo(p).Layer2.NumDroppedPacketsDueToRetries)
fprintf('\tThroughput: %d\n', numPayloadBits /
nodeInfo(p).Layer2.AverageRoundTripTime)
fprintf('\tLatency: %d\n',
nodeInfo(p).Layer2.AverageLatency)
end
```

